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Comparative Analysis of the Performance of Adjustable Windows in the Processingof Noisy Voice Signals

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ABSTRACT
Voice signal naturally is the most commonly used effective means of passing information or message for mankind. In the present stage of technological development voice signals are used to carry signal through transmissions channels to various locations of the globe. These voice signals experience interferences in these channels. The interfering components include additive white Gaussian noise, random noise, acoustic noise, electromagnetic noise, powerline noise and high frequency and low frequency noise components.These interferences corrupt the message contents of the voice signal if they are not removed or degraded. Different windows can be used to design a finite impulse response (FIR) filter for removing these noise components from the voice signals. In this paper five different windows, namely; Kaiser, single cosine term adjustable, double cosine term adjustable, HAT and HAS, windows are used individually to design a low pas FIR filter used in removing the high frequency noise components from a voice signal. A comparative analysis of the filters shows that Kaiser Window outperformed the other windows.

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I. INTRODUCTION

Using Voice signal is one of the effective means of passing information or message for mankind. Advancement in technology has provided the means to use voice signals to carry messages over transmission lines to very remote destinations. These transmission lines include mobile communication, telephone communication, multimedia communication and public address system transmission, lines. However, voice signals can experience interference from either the source or along this transmission lines by noise components such as additive white Gaussian noise [1], random noise, acoustic noise, electromagnetic noise, power line noise [2], high frequency and low frequency noise, and some other form of noise components.

For the integrity of the message in the voice signals not to be compromised before reaching the destination, any interfering signal must be removed. The aim of this paper is to remove low frequency noise from the voice signals using various adjustable windows. Normally the frequency range of voice signal is from about 200Hz to 3400Hz. Some researchers have used different windows to remove these noise components.

In [3] Pranab and Mohammand designed high pass filter with Kaiser window, which is an adjustable window, and some other windows for processing audio signals confined within voice signal frequency range. The cut off frequency for the high pass filter is 600Hz. Results indicate that the Kaiser Window filter provides sharp cut off for removing lower frequency components of voice signal. Sangeetha and Kannan [4] in using multirate signal processing for speech signals designed high pass FIR filters by using in addition to other windows, a Kaiser Window. The authors recorded and stored a voice signal of 8000Hz as a wave file for use in a matlab andadded additive white Gaussian noise (AWGN) to it. The designed filter is used to filter the signal and result shows that the filter provided a good performance.Saseendra and Rajesh [5] implemented a low pass filter with Kaiser Window. The sampling frequency is 40800Hz, cutoff frequency of 10800Hz and filter order is 20. Three different values of the window adjustment parameter **β equal**to **0.5, 3.5** and **8.5**are applied in sequence to filter high frequency noise from audio signals. Simulation results show that FIR filter with **β=8.5** gives best performance than the other values.In [6] Rajput and Bhadauria used a generalised adjustable window function to design low pass filter for filtering speech signals. The authors considered four different values of the adjustment parameter **α** including 0.71, 0.78, 0.5 and 0.54 with a sampling frequency of 16000Hz, filter order, 33 and cutoff frequency, 3200Hz. The result shows that each of the resulting windows is able to significantly filter out high frequency noise from the speech signal. Also in [7] Rajput and Bhadauria used anothergeneralised adjustable window function with adjustment parameter of **α**= 0.07 to design low pass filter for denoising speech signals of high frequency noise components. The sampling frequency is 8000Hz, filter order, 31 and cutoff frequency, 1200Hz. The result shows that the windowis able to significantly denoise the speech signal of high frequency noise components. In this paper five adjustable windows are used to remove high frequency noise from voice signal of dot windows media audio (.wma) format with a view to finding the best window for such processing.

II. ADJUSTABLE WINDOWS

Five adjustable windows are for consideration in this work. The windows are Kaiser, single cosine term generalised adjustable, double cosine termgeneralised adjustable, height adjustable triangular (HAT) and height adjustable sine (HAS), windows.

2.1. Kaiser Window

The Kaiser Window function isas in (1) [4, 8, 9, 10].

$$W_k(\beta, n) = \frac{J_o\left[\beta\left[1-\left(\frac{2n}{M-1}\right)^2\right]^{1/2}\right]}{J_o\beta} \qquad (1)$$

$$\text{Where} -\frac{M-1}{2} \le n \le \frac{M-1}{2} \quad \text{Where } J_o(P) = 1+\sum_{k=0}^{\infty}\left[(P/2)^k / k!\right]^2 \quad (2)$$

J_o (x) is the modified Bessel function of the first kind of order zero [4, 10]. M is the length of the window. **β**is a parameter that determines the shape of the window and can be selected independently. Fig. 1 shows the window when **β=16.**

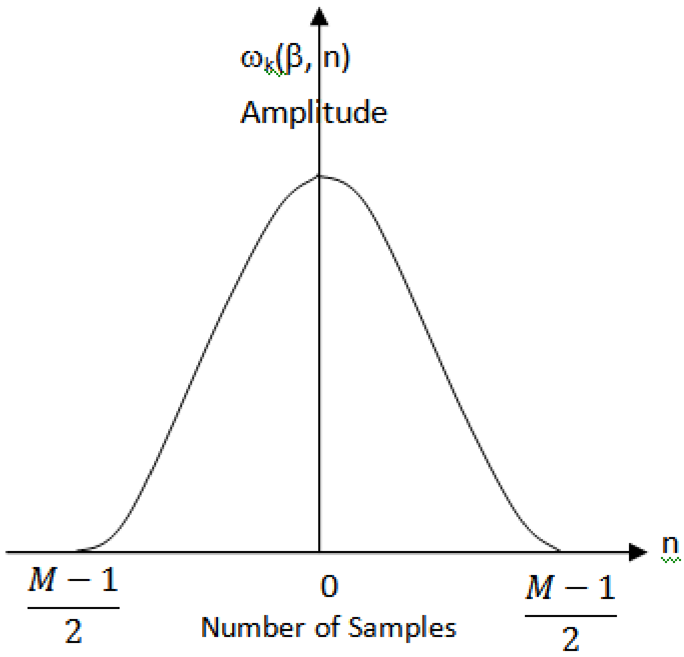


Fig. 1: Kaiser Window When **β=16**

2.2. Single Cosine Term Generalised Adjustable Window

The single cosine term generalised adjustable window is as in (2) [6]

$$w(n)=\alpha-(1-\alpha)\cos\left(\frac{2\pi n}{M-1}\right), 0 \le n \le M-1 \qquad (2)$$

where the adjustment parameter **α**varies between 0 and 1. Fig. 2 shows the window when **α =0.5.**

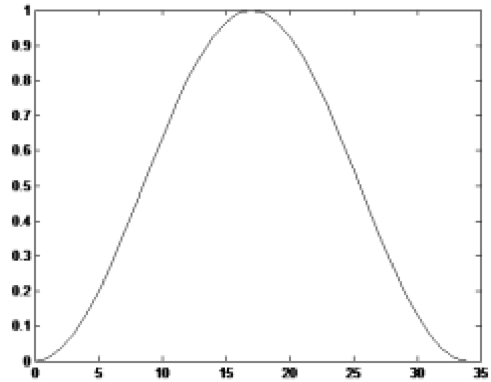


Fig. 2: Single Cosine Term Generalised Adjustable Window When **α =0.5**

2.3.Double Cosine Term Generalised AdjustableWindow

The double cosine term generalised adjustable window is as in (3) [7]

$$w(n)=\frac{1-\alpha}{2}-0.5\cos\left(\frac{2\pi n}{M-1}\right)+\frac{\alpha}{2}\cos\left(\frac{4\pi n}{M-1}\right), 0 \le n \le M-1 \qquad (3)$$

where the adjustment parameter **α**varies between 0 and 1. Fig. 3 shows the window when **α =0.2**

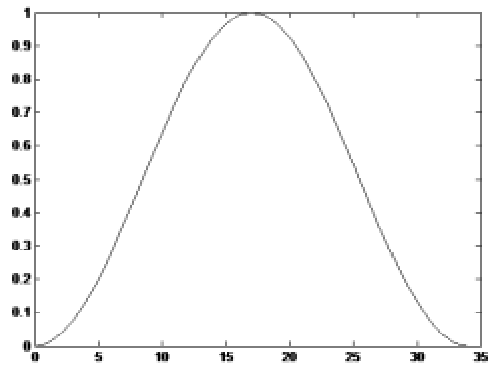


Fig.3: Double Cosine Term Generalised Adjustable Window When **α =0.2**

2.4.HAT Window

The HAT window function is as in (4) [10, 11]

$$w(n)=\begin{cases}\alpha+(2-2\alpha)n/(M-1), & 0 \le n \le \frac{M-1}{2}\\ 2-[\alpha+(2-2\alpha)n/(M-1)], & \frac{M-1}{2} \le n \le M-1\end{cases} \qquad (4)$$

wherethe adjustment parameterα varies from 0 to 1 and adjusts the amplitude of the window. Fig.4 is the HAT window when **α=0.1**

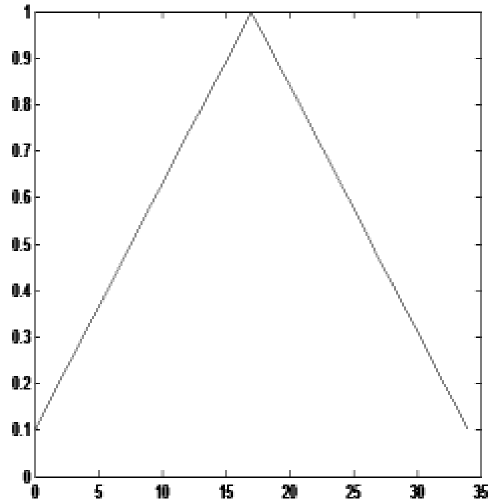


Fig.4: HAT window when **α=0.1**

2.5. HAS Window

The HAS window function is as in (5) [10, 12]

$$w(n) = \begin{bmatrix} \alpha + \sin\left[\dfrac{2\sin^{-1}(1-\alpha)}{L} n\right], 0 \le n \le \dfrac{M-1}{2} \\ \alpha + \sin\left[\dfrac{(L-n)2\sin^{-1}(1-\alpha)}{L}\right], \dfrac{M-1}{2} \le n \le M-1 \end{bmatrix} \quad (5)$$

wherethe adjustment parameter**α** varies from 0 to 1 and adjusts the amplitude of the window. M is the length of the window and L=M-1.Fig.5 is the HAS window when **α=0.04.**

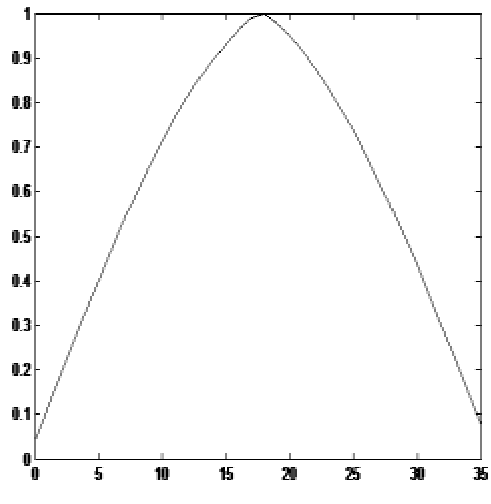


Fig.5: HAS window when **α=0.04**

III. DESIGN OF LOW PASS DIGITAL FIR FILTER

In this design the five adjustable windows are used to design a low pass FIR filter individually. The filter length is 35, that is, order 34, cutoff frequency, 3200Hz and sampling frequency, 44100Hz,the impulse, magnitude and phase responses of the filter for each window are obtained and are depicted below.

3.1. Kaiser Window Responses

The impulse, magnitude and phase responses when the adjustment parameter is **β=16** are depicted in fig6a, fig.6b and fig.6c respectively.

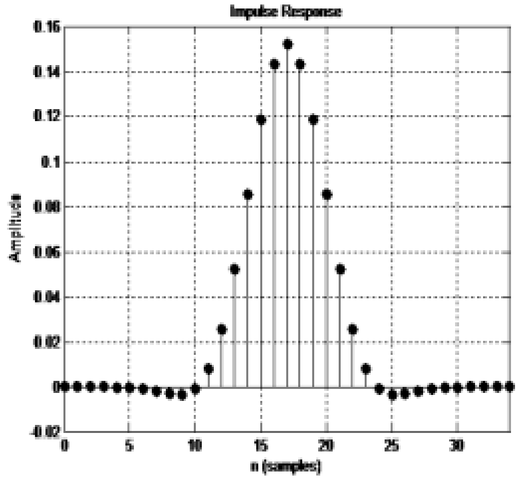


Fig.6a: Impulse Response When **β=16**

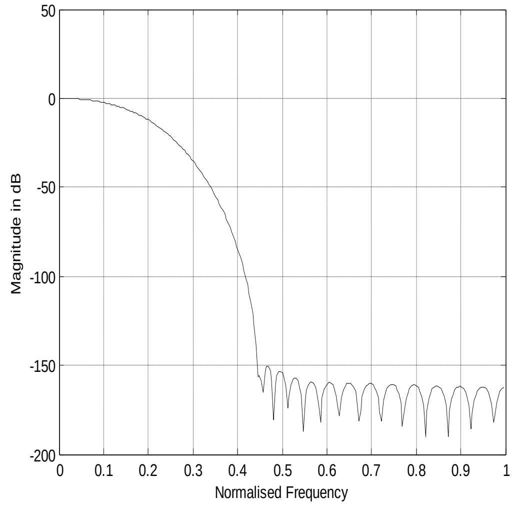
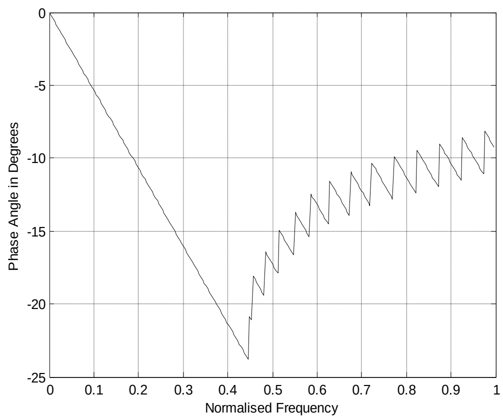


Fig.6b: Magnitude Response When **β=16**



.**Fig.6c**: Phase Response When **β=16**

3.2. Single Cosine Term Generalised
Adjustable Window Responses

The impulse, magnitude and phase responses when the adjustment parameter is **α =0.5** are depicted in fig7a, fig.7b and fig.7c respectively.

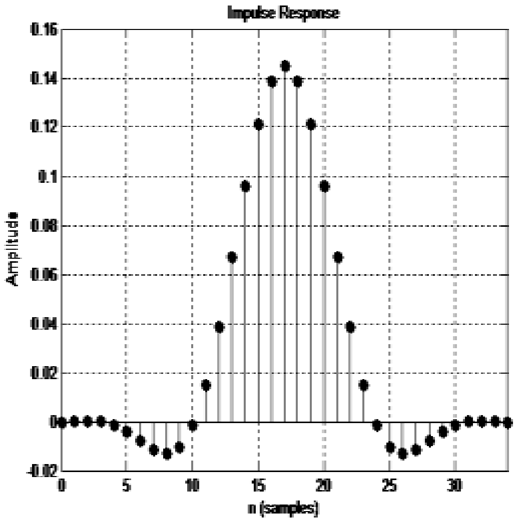


Fig.7a: Impulse Response When **α =0.5**

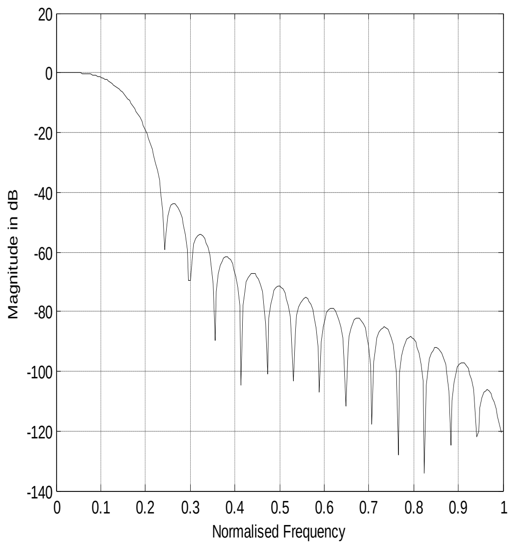


Fig.7b: Magnitude Response When **α =0.5**

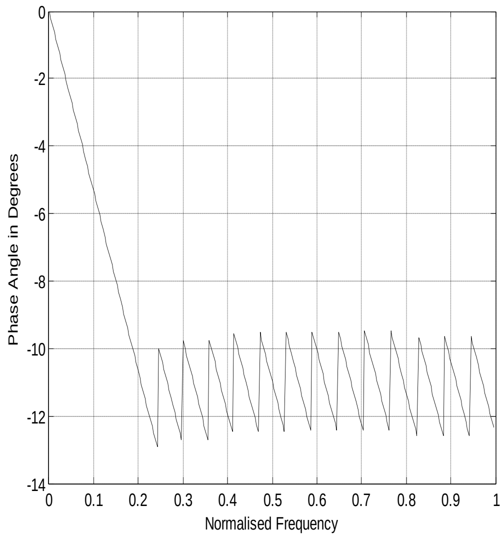


Fig.7a: Phase Response When **α =0.5**

3.3. Double Cosine Term Generalised Adjustable Window Responses

The impulse, magnitude and phase responses when the adjustment parameter is **α =0.2** are depicted in fig8a, fig.8b and fig.8c respectively.

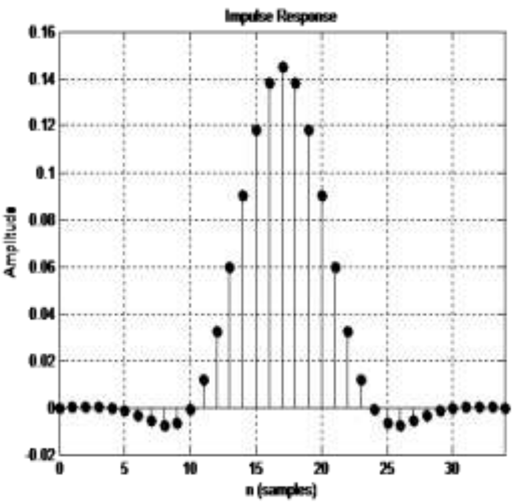


Fig.8a: Impulse Response When **α =0.2**

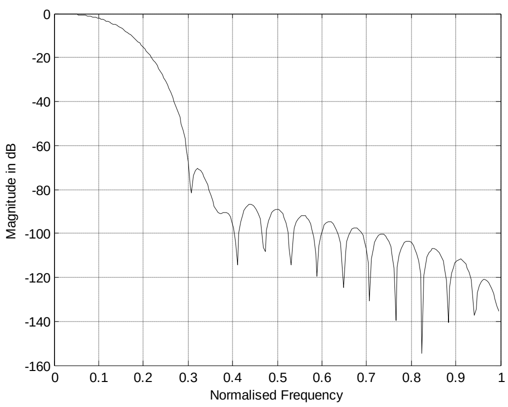


Fig.8b: Magnitude Response When **α =0.2**

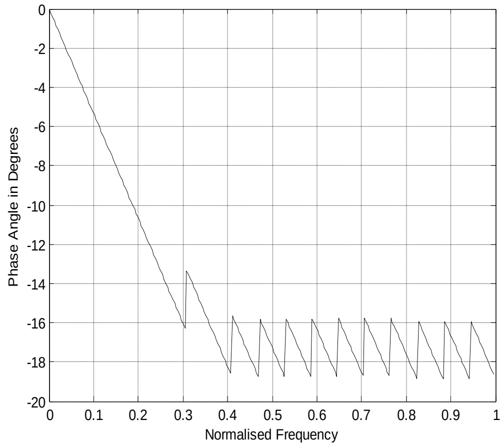


Fig.8c: Phase Response When **α =0.**

3.4. HAT Window Responses

The impulse, magnitude and phase responses when the adjustment parameter is **α =0.1**are depicted in fig9a, fig.9b and fig.9c respectively.

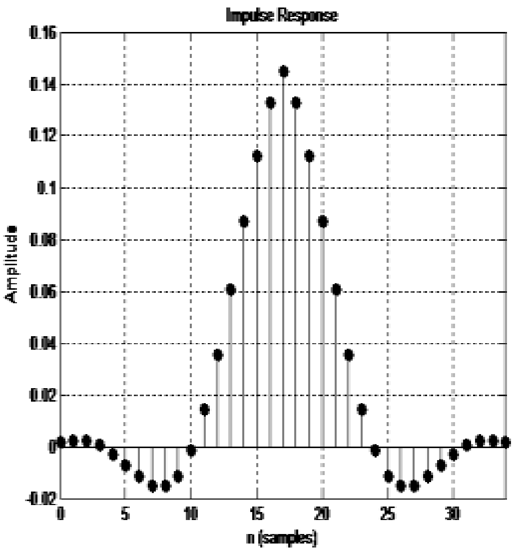


Fig.9a: Impulse Response When **α =0.1**

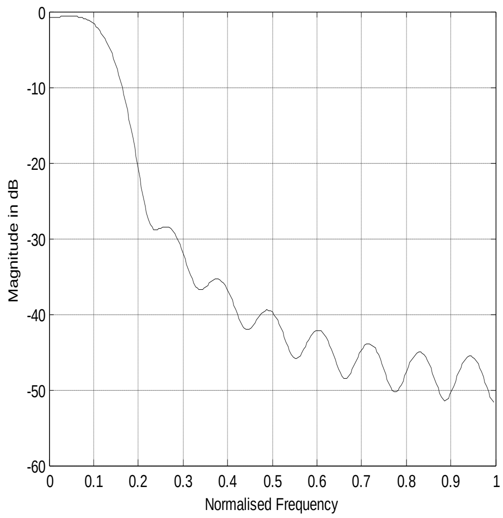


Fig.9b: Magnitude Response When **α =0.1**

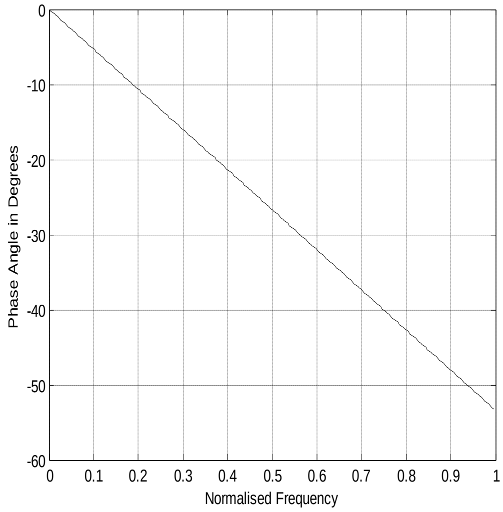


Fig.9c: Phase Response When **α =0.1**

3.5. HAS Window Responses

The impulse, magnitude and phase responses when the adjustment parameter is **α =0.04** are depicted in fig10a, fig.10b and fig.10c respectively.

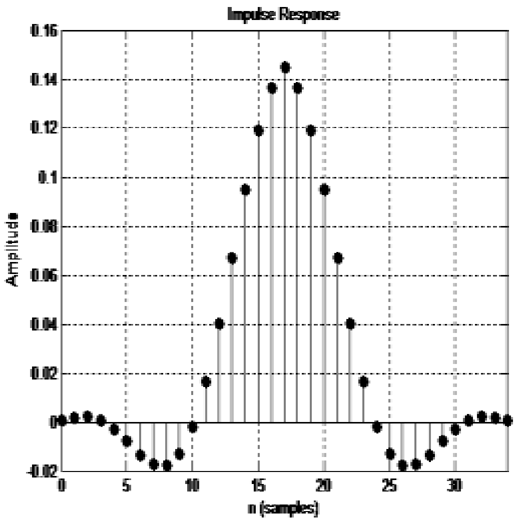


Fig.10a: Impulse Response When **α =0.04**

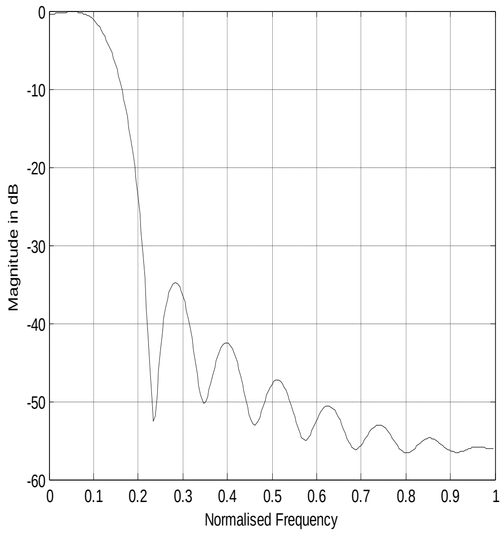


Fig.10b: Magnitude Response When **α=0.04**

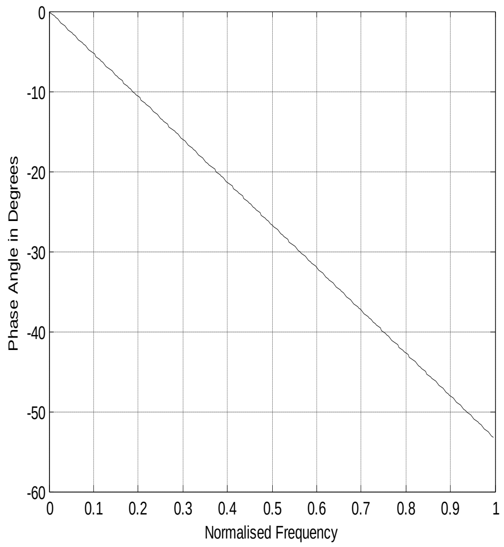


Fig.10c: Phase Response When **α =0.04**

IV. RESULTS

A real voice signal is generated by a voice statement captured with a microphone and recorded in a file of a computer system in dot windows media audio (.wma) format. The length of the recorded voice signal is 466887. The signal is loaded into a matlabworkspace using "audioread" instruction. In the workspace a noise component of 4500Hz and above is generated with matlab and mixed with the voicesignal to constitute a contaminated voice signal. The noise free voice signal is shown in fig.11 while fig.12 depicts the noise component, and the contaminated voice signal depicted in fig.13. The contaminated voice signal is applied separately to the filters designed with each of the windows and the outputs recorded. Figures 14, 15, 16, 17 and 18 show the recorded output of the filters. From the noise free voice signal of fig.11, the contaminated voice signal of fig.13 and the filtered voice signals of fig14 to fig.18 it is clear that each window reduced the noise in the contaminated voicesignal. The window with the best performance will be determined from the spectral densities of the filtered signals.

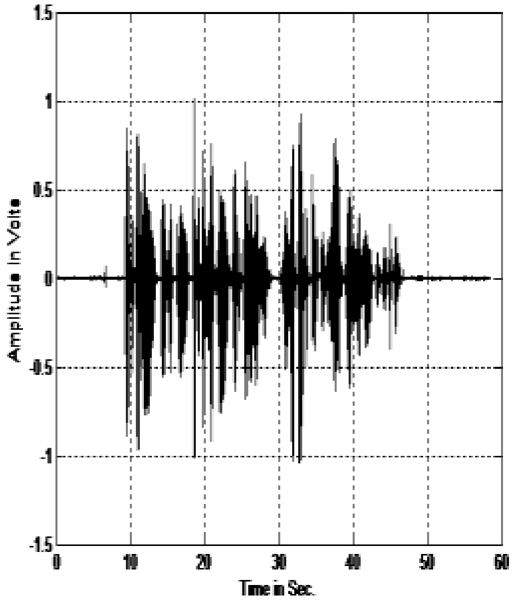


Fig.11: Noise Free Voice Signal

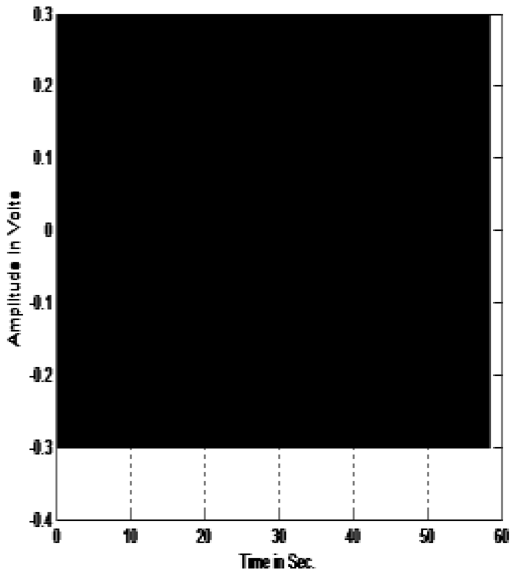


Fig.12: Noise Signal

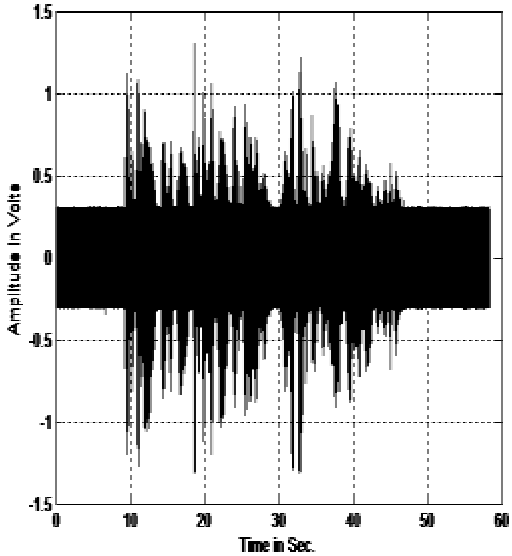


Fig.13: Contaminated Voice Signal

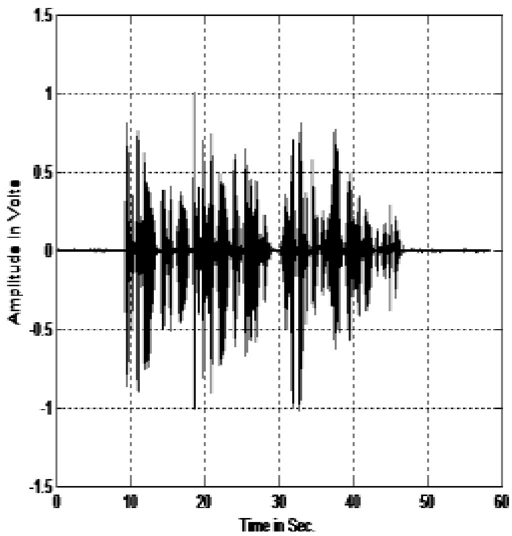


Fig.14: Voice Signal Filtered With Kaiser Window

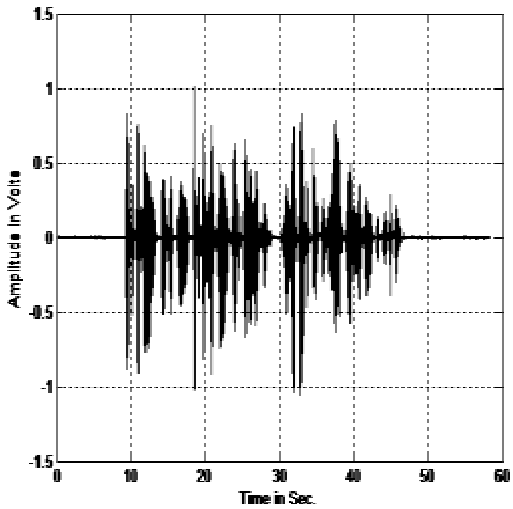


Fig.15: Voice Signal Filtered With Single Cosine Term Generalised Adjustable Window

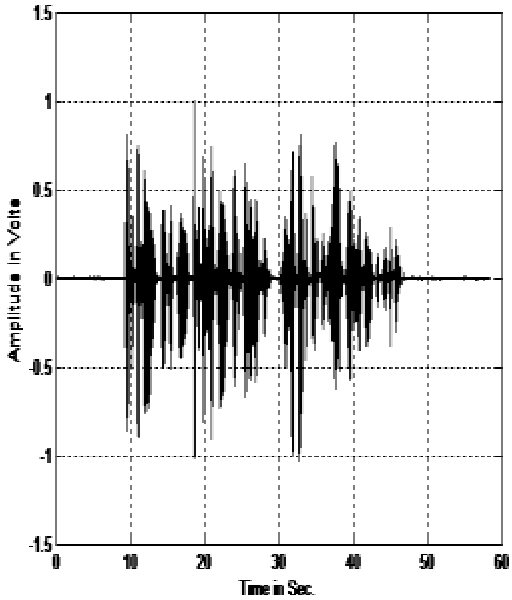


Fig.16: Voice Signal Filtered With Double Cosine Term Generalised Adjustable Window

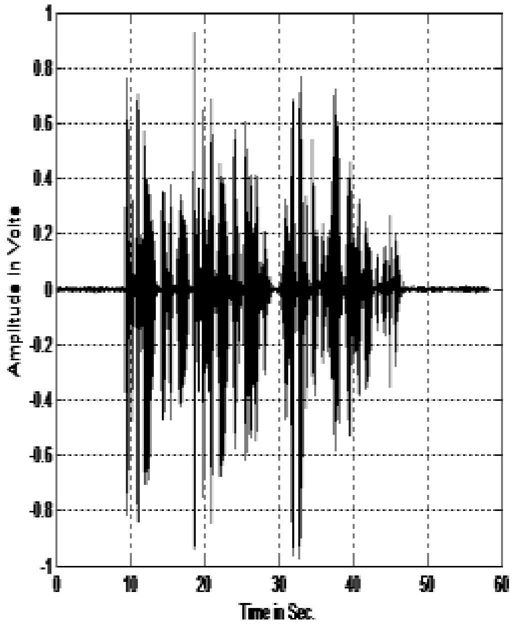


Fig.17: Voice Signal Filtered With HAT Window

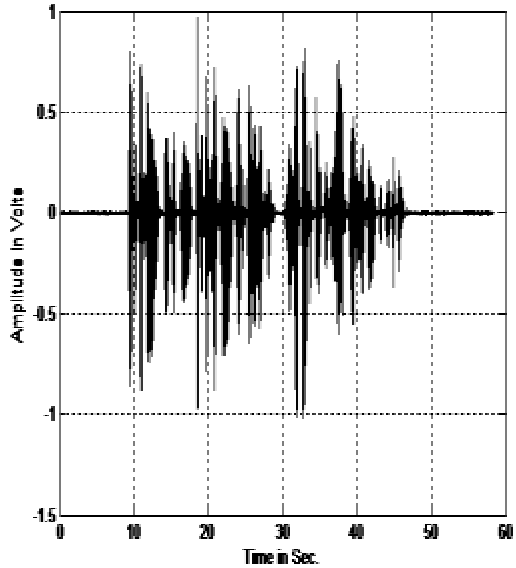


Fig.18: Voice Signal Filtered With HAS Window

4.1. Signal Power Level

The window that provides the optimum noise attenuation can be determined from the comparative analysis of the power spectral densities of the filtered voice signals [10, 13, 14]. Fig. 19 is the power spectral density of a noise free voice signal while the spectral density of the contaminated voice signal is depicted in fig. 20. Fig.21 to fig.25 is the power spectral densities of the filtered voice signal with each of the windows**.** From fig. 13the noise component in the contaminated voice signal is much more than other frequency points at normalised frequency of 0.875. Therefore effectiveness of each window is determined at this frequency point. Table 1 below shows the summary of the power levels in dB of the filtered signals by each window**.**

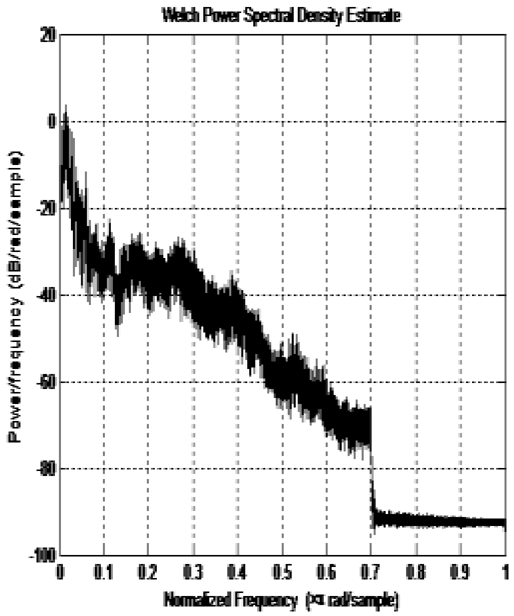


Fig.19: Power Spectral Density of Noise Free Voice Signal

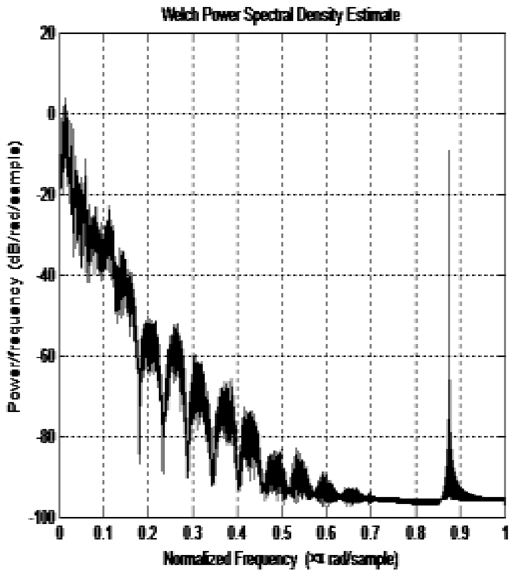


Fig.20: Power Spectral Density of Contaminated Voice Signal

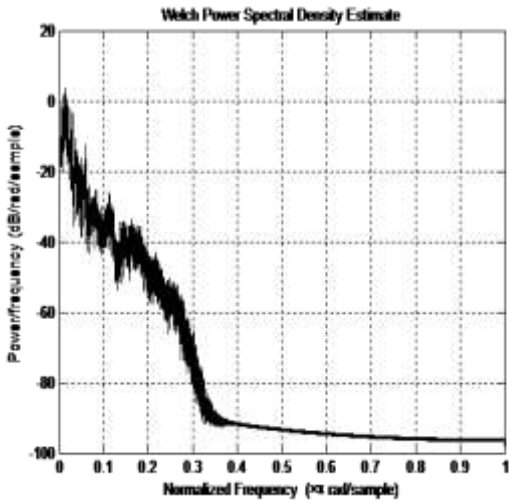


Fig.21: Power Spectral Density of Voice Signal Filtered With Kaiser Window

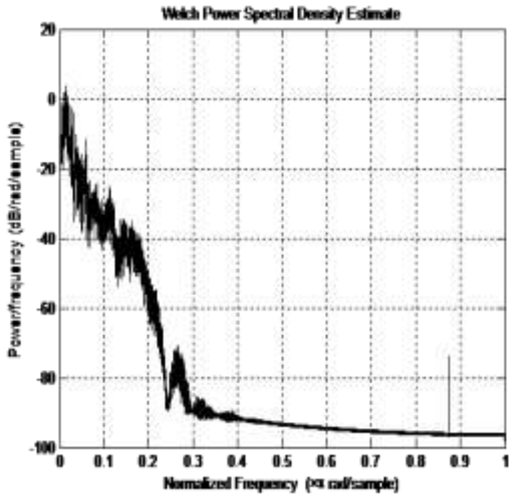


Fig.22: Power Spectral Density of Voice Signal Filtered With Single Cosine Term Generalised Adjustable Window

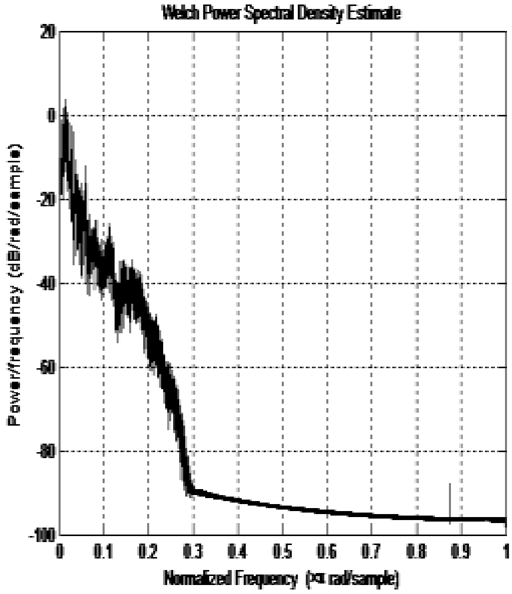


Fig.23: Power Spectral Density of Voice Signal Filtered With Double Cosine Term Generalised Adjustable Window

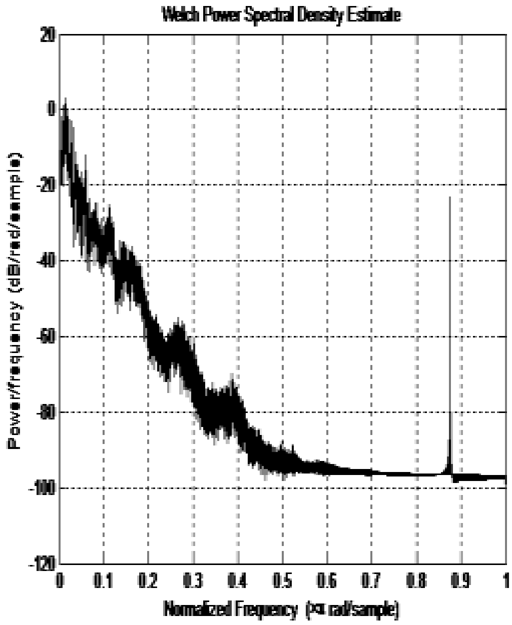


Fig.24: Power Spectral Density of Voice Signal Filtered HAT Window

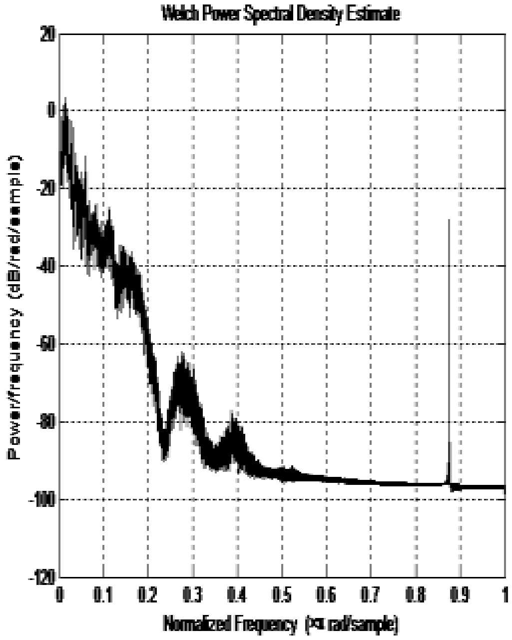


Fig.25: Power Spectral Density of Voice Signal Filtered HAS Window

Table 1: Power Levels of Voice Filtered Signals at 0.875 Normalised Frequency

Windows	Filtered voice power levels
Kaiser Window	-92.99dB
Single cosine term generalized adjustable window	-74.98dB
Double cosine term generalised adjustable window	-89.87dB
HAT window	-22.78dB
HAS window	-28.02dB

From fig.19 the power level of the real noise free voice signal is -90.98dB and from fig.20 the power level of the contaminated voice signal is +28.15dB. This means that the noise component added a lot of noise power to the noise free voice signal to the tune of 28.15-(-90.98)=119.13dB. From table 1 it can be seen that the best performing window is Kaiser Window with attenuation level of -92.99dB, followed by double cosine term generalised adjustable window with attenuation level of -89.87dB.

V. CONCLUSION

It can be concluded that the five windows can denoise voice signal of high frequency noise component but the Kaiser window is the most effective of them because it does not only completely remove the noise components but does not also distort the signal or exhibit phase non linearity.

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